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## Description

# DIRECTIONAL MICROPHONE SYSTEM

## Technical Field

This invention relates to microphones and, in particular, to directional microphones.

## Background Art

Advancements in the computer hardware and software industry are occurring rapidly in a variety of areas. One such area of advancement has been in the field of voice applications for personal computers. Voice applications, such as speech recognition, allow the user of a personal computer to communicate, or to interface, with the machine and the software in a simple and effective manner by simply speaking to the machine. International Business Machine Corporation's ViaVoice speech dictation product provides an example of this type of application. Other voice applications that utilize technology recently developed in this area include computer telephony, internet telephony and audio / video conferencing applications.

As the development and use of these voice applications has grown in the personal computer marketplace, so has the need for a hands-free or farfield microphone product that is capable of effectively receiving voice commands, instructions or speech, while doing so in a cost efficient manner. While the issue of cost for such a microphone product by itself presents a formidable obstacle to a successful microphone system, another recognized problem has emerged. That problem involves the pickup of reverberation as well as noise from sources other than the intended sound source, which is the talker. Noise and reverberation pickup lessens the overall performance of the speech application.

For example, a personal computer generally includes a fan as one of the integrated parts that is necessary for the computer to function properly by cooling the internal components of the computer. Unfortunately, such fans in operation create a certain amount of noise. Other integrated parts of a personal computer which can create unwanted noise include disk drives and CD-ROM drives. Additionally, personal computers can be operated in noisy environments, such as an office setting. Specifically, in an office setting the user of a computer-related voice application may experience unwanted noise generated by sources such as other individuals in the workplace, office equipment (e.g., telephones and photocopy machines) and other types of

undesired background noise. Such unwanted noise can directly decrease the effectiveness of microphones used in association with voice applications for personal computers.

Other devices utilizing microphones have some or all of the same issues in dealing with unwanted noise. These devices include, but are not limited to, a headset, a telephone handset, a speaker phone (hands-free telephone product), or a video camera. Microphones used in each of these devices are exposed to unwanted noise that is present in the surrounding environments of their respective use. For example, a microphone in a speaker phone application which is utilized in an office setting can be exposed to undesired background noise, such as noise produced by nearby office equipment, which can decrease the sound quality of the product. Because the environment associated with personal computer applications is illustrative of a typical acoustic environment, it is generally referred to herein for discussion purposes.

One attempt to solve the problem of noise pickup in the field of hands-free microphone products for use with personal computers and related applications has been to utilize directional microphone systems. For example, a directional microphone system can be formed through the use of a gradient microphone element in which a time varying sound pressure is exerted on two sides of a diaphragm forming a pressure gradient response and, ultimately, a microphone with a directional response. Unfortunately, this type of microphone system is usually limited in its effectiveness. Specifically, this type of microphone system, which has a "fixed" beam response or pickup pattern, is mainly effective at reducing noise pickup when the noise source radiates sound from a direction of low microphone sensitivity, also known as the "null" angle. However, because the specific location of unwanted noise sources is often difficult, if not impossible, to anticipate or control, the microphone does not effectively diminish the pickup of noise and the microphone's performance is significantly decreased. Additionally, such single element microphone systems are characterized by fairly broad beam patterns which result in limited noise and reverberation reduction.

It is also possible to generate a gradient microphone response by utilizing two or more omnidirectional microphone elements. This approach requires the use of an electrical subtraction. Additionally, this approach requires the use of "matched" microphone elements, i.e., microphones, that have the same, or closely similar, sensitivity response. Again, however, the identification of these "matched" elements has resulted in an increased cost for the system. Also,

this type of microphone system still suffers from the performance issues that are associated with fixed beam microphone systems.

Other devices in the prior art, such as those disclosed in U.S. Pat. Nos. 4,802,227 and 5,699,437, generally disclose microphone systems with multiple omnidirectional microphone elements that attempt to reduce unwanted noise pickup and can be used in conjunction with personal computers. These systems attempt to adapt the directional response of the system to changes in the locations of the primary sound source and the noise sources by the use of complex algorithms. Thus, these microphone systems are unnecessarily complex, and they still exhibit some shortcomings with regard to performance and cost as discussed above in association with less complicated microphone systems.

One attempt to address some of the problems associated with the movement of noise sources in relation to fixed beam microphone products has been to utilize microphone systems that exhibit adaptive, although in a limited fashion, response characteristics. An example of an adaptive system is found in U.S. Pat. No. 5,473,701. This solution is limited, however, in that it restricts the region where the direction of lowest microphone sensitivity, again known as the "null" angle, can occur. Additionally, this microphone system restricts the region from which the desired sound, generally the person talking, can be generated. Thus, while the limited adaptive characteristics exhibited by this system are somewhat beneficial, they do not provide a complete solution to the special needs associated with the use of microphones with voice applications for personal computers, and the other microphone system applications generally described previously.

In addition to the above-referenced performance shortcomings of directional microphone systems existing in the prior art, another disadvantage is associated with products currently in use. Specifically, directional microphones imbedded into products generally limit the frequency bandwidth of the microphone system. While a restricted frequency bandwidth is reasonable for some telephony applications, it is undesirable for newer speech applications. Thus, current directional microphone systems have not addressed the need for an extended frequency bandwidth response.

By way of example of such directional microphone systems in the prior art, assume "d" is the distance between first and second sound inlets in a gradient microphone system. That

distance "d" determines the output and frequency response of the overall microphone system. Increasing "d" results in an increase in the output, and the electrical signal-to-noise ratio, but also decreases the frequency bandwidth of the system. The system of U.S. Pat. No. 5,226,076 is one example of a prior art device where acoustic tubes are used to increase "d" to obtain greater sensitivity at the expense of reducing overall bandwidth.

The "d" commonly employed by prior art multiple microphone systems is approximately two and one-half (2.5) centimeters because that distance results in what has become to be considered an acceptable level of microphone sensitivity. However, this commonly-used "d" results in an upper cutoff frequency of only approximately five (5) kilohertz (kHz). Some voice applications including, but not limited to, those used in conjunction with personal computers, for example voice recognition software, would be better served by an extended bandwidth. Thus, it is desired that a microphone system produce a directional response pattern and also produce a broader frequency range, yet still realize the sensitivity increase associated with a larger "d".

The prior art also reveals efforts to address some of the performance and cost issues associated with other microphone systems. Specifically, other microphone systems involve the use of expensive and complicated dedicated analog or digital circuits for signal processing. For example, PictureTel Corporation's Virtuoso™ and LimeLight™ products each utilize dedicated digital signal processor integrated circuits to decrease the pickup of undesired noise. While this type of system demonstrates some improvement in performance characteristics over other standard directional microphones, such as those described above, these microphone systems suffer from a cost disadvantage as they are expensive to manufacture. Thus, these systems may not provide a cost effective solution to the need for a reasonably priced microphone system for use with, by way of example, voice applications in the personal computer market.

Another example of a microphone system that utilizes its own built-in electronics is Telex Communications, Inc.'s Aria™ Desktop Dictation Microphone. While this system may also demonstrate some improvement in performance characteristics over other standard directional microphones, the use of dedicated circuits causes this system to be overly complex and expensive to manufacture.

Yet another type of microphone system currently in use is a microphone headset apparatus. This type of system is used because unwanted noise cancellation can be best achieved

in some current microphone products only if the microphone is placed less than one and one-quarter (1.25) centimeters from the speaker's mouth. The only practical way to achieve such a short distance has been to utilize a microphone headset. Such headset systems are offered by Andrea Electronics Corporation, and, for example, one such product is disclosed in U.S. Pat. No. 5,251,263. However, headset-type systems have several disadvantages from the user's point of view. Specifically, many users find such headsets uncomfortable to wear. Additionally, when headset microphone systems are utilized in conjunction with computer applications, they have the undesirable effect of "tethering" the user to the computer. Many users find this to be a significant disadvantage, as it is desirable to be able to move freely about a work area. Furthermore, the system disclosed in U.S. Pat. No. 5,251,263 contains a dedicated processor and an analog-to-digital and digital-to-analog interface. These additional components increase the complexity and cost of this system. Thus, the system covered by that patent is relatively expensive to produce and maintain.

A still further example of an attempt to cancel the undesired pickup of noise in a microphone system is described in U.S. Pat. No. 5,825,897. However, the method described in that patent requires the production of an anti-noise signal in order to cancel undesired noise. Again, this type of system requires the use of additional components that may be expensive to produce and maintain.

**Disclosure of Invention**

The present invention comprises a directional microphone system with multiple microphone elements which are combined in a manner to produce a monaural signal with an extended frequency bandwidth response. In one embodiment, the directional microphone system is a stand-alone, farfield system which comprises a first and second microphone means, a means for combining signals produced by the microphone means and a cross-over means for producing a monaural signal.

The first and second microphone means each receive acoustic energy composed of sound from a desired source and sound from undesired background noise. First and second microphone means convert the acoustic energy into first and second signals, respectively. Next, the combining means combines the first and second signals into a combined signal.

The combined signal is then provided to the cross-over means. The cross-over means comprises a first filter means for filtering one of the first or second signals and a second filter means for filtering the combined signal. In this embodiment, the first filter means comprises a high band equalization and filtering delay and a low band equalization and filtering delay. Also, the second filter means comprises a mid band equalization and filtering delay. The filtering delay is used to time-align the first and second filter means and to avoid cancellation at the edges of the frequency bands. The cross-over means also comprises a unifying means for combining each signal processed by the first and second filter means, respectively. Thus, the cross-over means produces a monaural signal representing substantially the sound produced by the desired source, the monaural signal having an extended frequency bandwidth response. Additionally, the invention provides the advantage of not requiring complex, dedicated circuitry or expensive, matched microphone elements.

In another form of the invention, the directional microphone system is used in connection with a computer, the computer comprising a processor and first and second stereo inputs. In this form, the first and second microphone means of the invention are connected to the first and second stereo inputs of the computer. In yet another form of the present invention, the first filter means comprises a high-pass filter, while the second filter means comprises a low-pass filter.

The present invention provides the advantage of utilizing multiple microphone elements to produce a monaural signal with an extended frequency bandwidth response that substantially represents the sound produced by a desired source while substantially reducing the pickup of undesired background noise. Additionally, the present invention provides a cost effective system which does not require the use of complex, dedicated circuitry or matched microphone elements. Furthermore, the present invention provides a system which can be utilized in connection with a computer by utilizing the stereo inputs and processing capabilities of the computer.

## Brief Description of the Drawings

FIG. 1 shows a block diagram of one embodiment of the directional microphone system of the present invention.

FIG. 2 shows a graphical representation of the extended frequency bandwidth response of the embodiment of FIG. 1.

FIG. 3 shows a block diagram of a second embodiment of the directional microphone system of the present invention.

FIG. 4 shows a block diagram of the cross-over portion of the directional microphone system of FIG. 3.

**Best Mode for Carrying Out the Invention**

Referring to FIG. 1, there is shown a block diagram of one embodiment of the directional microphone system according to the present invention, generally designated 10, which produces a monaural signal substantially representative of sound from a desired sound source. Directional microphone system 10 includes first and second microphone elements 14, 16, respectively, with each of microphone elements 14, 16, respectively, characterized by a certain sensitivity in its response. As shown in FIG. 1, first and second microphone elements 14, 16, respectively, are connected to a personal computer 12. While this embodiment is used in connection with a personal computer, it should be understood that this embodiment of directional microphone system 10 is a stand-alone, farfield microphone system. In this embodiment of the invention, first and second microphone elements 14, 16, respectively, are each first-order gradient microphone elements, such as EM118 microphones manufactured by Primo Microphones, Ltd. of Japan. Additionally, first and second microphone elements 14, 16, respectively, are arranged in close proximity to each other, with spacing for sufficient sensitivity. As such, directional microphone system 10 provides the advantage of being compact in size.

Also as shown in FIG. 1, directional microphone system 10 includes matching algorithm 18 of the type used to match the sensitivities of first and second microphone elements 14, 16, respectively. Additionally, directional microphone system 10 includes pattern former 20. Directional microphone system 10 also includes cross-over portion 22, which comprises a first filter portion 24 and a second filter portion 26. Matching algorithm 18, pattern former 20, cross-over portion 22, first filter portion 24 and second filter portion 26 are discussed in greater detail later herein.

In operation, first and second microphone elements 14, 16, respectively, each receive acoustic energy composed of sound from a desired source, such as speech, and undesired background noise, such as noise produced from nearby office machines. First microphone element 14 converts the acoustic energy into a first electrical output signal, generally designated

28, and second microphone element 16 converts the acoustic energy into a second electrical output signal, generally designated 30. First and second microphone elements 14, 16, respectively, are connected by first and second connectors 32, 34, respectively, to first and second stereo inputs 36, 38, respectively, of personal computer 12. First and second stereo inputs 36, 38, respectively, are connected by third and fourth connectors 40, 42, respectively to preamplifier 44. Preamplifier 44 provides gain to first and second electrical output signals 28, 30, respectively, producing first and second preamplifier output signals 46, 48, respectively.

First and second preamplifier output signals 46, 48, respectively, are connected by fifth and sixth connectors 50, 52, respectively, to a standard analog-to-digital (A/D) converter 54. A/D converter 54 converts first and second preamplifier output signals 46, 48, respectively, to first and second digital signals 56, 58, respectively. The sampling rate utilized by A/D converter 54 in the present embodiment is 44.1 ks/sec.

In the present embodiment, first and second digital signals 56, 58, respectively, are connected by seventh and eighth connectors 60, 62, respectively, to processor 64, which in the present embodiment, is also part of personal computer 12. In the present embodiment, processor 64 controls microphone element matching algorithm 18. It will be appreciated by those skilled in the art that in other embodiments of the present invention, processor 64 could also implement matching algorithm 18.

In the present embodiment of directional microphone system 10, first and second digital signals 56, 58, respectively, are provided to matching algorithm 18. Matching algorithm 18 matches the sensitivities of first and second microphone elements 14, 16, respectively, to improve the directional characteristics of directional microphone system 10. In operation, matching algorithm 18 produces first and second gain corrected signals 66, 68, respectively. It will be appreciated by those skilled in the art that the use of matching algorithm 18 provides an inexpensive method to match the sensitivities of first and second microphone elements 14, 16, respectively. Additionally, the use of matching algorithm 18 prevents the need to use expensive dedicated circuits to achieve a similar result. It will be appreciated by those skilled in the art that any one of a variety of algorithms could be used to achieve the result of matching algorithm 18.

First gain corrected signal 66 is provided to first filter portion 24 of cross-over portion 22. In the present embodiment, first filter portion 24 includes high band equalization and

filtering delay 70 and low band equalization and filtering delay 72. First gain corrected signal 66 is provided to each of high and low band equalization and filtering delays 70, 72, respectively. High band equalization and filtering delay 70 produces high band signal 74. Low band equalization and filtering delay 72 produces low band signal 76. Thus, in the present embodiment, the output of first filter portion is comprised of high band signal 74 and low band signal 76. It will be appreciated by those skilled in the art that second gain corrected signal 68 could be provided to first filter portion 24, instead of first gain corrected signal 66.

First and second gain corrected signals 66, 68, respectively, are provided to pattern former 20. In the present embodiment, pattern former 20 performs a subtraction to combine first and second gain corrected signals 66, 68, respectively, to produce combined signal 78. It will be appreciated by those skilled in the art that pattern former 20 as described herein is only one example of a method that could be used to combine first and second gain corrected signals 66, 68, respectively. For example, pattern former 20 could comprise a delay and subtraction technique. Thus, it should be understood that other forms of pattern former 20 are possible and remain within the scope of the present invention.

Combined signal 78 is provided to second filter portion 26. In the present embodiment, second filter portion 26 is comprised of a mid band equalization and filtering delay. Second filter portion 26 produces mid band signal 80.

Each of high, low and mid band signals 74, 76, 80, respectively, are provided to first algebraic summing unit 82. First algebraic summing unit 82 unifies high, low and mid band signals 74, 76, 80, respectively, by, in the present embodiment, simply adding them together to produce monaural signal 84. It will be appreciated by those skilled in the art that this unification could be completed by other types of devices and still be within the scope of this invention.

As a result of the unification of filtered signals from one microphone element and a combination of both microphone elements, directional microphone system 10 provides the advantage of producing an extended frequency bandwidth response. Furthermore, because the present embodiment utilizes high and low band equalization and filtering delays 70, 72, respectively, along with the mid band equalization and filtering delay of second filter portion 26, the bandwidth of monaural signal 84 is extended at both the high and low ends. Such extension

of the frequency bandwidth response of the present embodiment of directional microphone system 10 is graphically depicted in FIG. 2.

Referring further to FIG. 1, in the present embodiment of directional microphone system 10, first constant 86 is provided to first multiplier 88. Also, first digital signal 56 is provided to first multiplier 88. First multiplier 88 produces first passthrough output 92. First constant 86 is provided to second multiplier 90, as is second digital signal 58. Second multiplier 90 produces second passthrough output 94.

Monaural signal 84 is provided to second algebraic summing unit 96. First passthrough output 92 is also provided to second algebraic summing unit 96. Second algebraic summing unit 96 produces first channel output 100. Monaural signal 84 is also provided to third algebraic summing unit 98. Second passthrough output 94 is additionally provided to third algebraic summing unit 98. Third algebraic summing unit 98 produces second channel output 102.

Assuming a user desires to have the present embodiment of directional microphone system 10 operate to produce monaural signal 84 which effectively reduces the pick-up of undesired background noise and produces an extended frequency bandwidth response, then the value of first constant 86 would be set to equal zero (0). As a result, first and second multipliers 88, 90, respectively, will produce first and second passthrough outputs 92, 94, respectively, each equal to zero (0). With first and second passthrough outputs 92, 94, respectively, equal to zero (0), second and third algebraic summing units 96, 98, respectively, will produce identical first and second channel outputs 100, 102, each equal to monaural signal 84.

Alternatively, by utilizing first and second multipliers 88, 90, respectively, and second and third algebraic summing units 96, 98, respectively, directional microphone system 10 provides a capability for bypassing the part of the system that produces monaural signal 84.

For example, suppose a user desires first and second channel outputs 100, 102, respectively, to merely equal first and second digital signals 56, 58, respectively. In order to achieve such a result, each of high, low and mid band signals 74, 76, 80, respectively, must all equal zero (0). It will be appreciated by those skilled in the art that setting each of high, low and mid band signals 74, 76, 80, respectively, equal to zero (0) could be achieved in a varierty of ways. For example, constants which are set equal to zero (0) could be provided to each of first

and second filter portions 24, 26, which would in turn cause each of high, low and mid band signals 74, 76, 80, respectively, to be equal to zero (0).

Because each of high, low and mid band signals 74, 76, 80, respectively, would be equal to zero (0), so would monaural signal 84. Thus, second and third algebraic summing units 96, 98, respectively, would add the values of monaural signal 84 (equal to zero) to first and second passthrough outputs 92, 94, respectively. First and second channel outputs 100, 102, respectively, would therefore simply equal first and second passthrough outputs 92, 94, respectively. If first constant 86 is set to equal one (1), then first and second multipliers 88, 90, respectively, produce first and second passthrough outputs 92, 94, respectively, equal to first and second digital signals 56, 58, respectively.

Referring now to FIG. 3, there is shown an alternative embodiment of directional microphone system 10. The embodiment shown in FIG. 3 is also used in connection with personal computer 12. Unlike the prior embodiment, however, the present embodiment is designed to be integrated into personal computer 12. The present embodiment of directional microphone system 10 includes first and second microphone elements 14, 16, respectively, each of these microphone elements characterized by a certain sensitivity in its response. In this embodiment, first and second microphone elements 14, 16, respectively, are each omnidirectional microphone elements, such as model EM100T microphones manufactured by Primo Microphones, Ltd., of Japan. These omnidirectional microphone elements provide the advantage of being relatively inexpensive. Additionally, first and second microphone elements 14, 16, respectively, are arranged in close proximity to each other, with spacing for sufficient sensitivity. Similar to the prior embodiment, the present embodiment of directional microphone system 10 includes matching algorithm 18.

Also as shown in FIG. 3, the present embodiment includes cross-over portion 22, which comprises first and second filter portions 24, 26, respectively. Additionally, the present embodiment also includes adaptive algorithm 104. Adaptive algorithm 104 produces combination signal 106, which is characterized by a direction of low sensitivity. It is noted that elements 28 through 64 are also shown in FIG. 3. Further discussion is not provided regarding those elements, however, as they have been previously described in the discussion regarding FIG. 1.

Referring further to FIG. 3, as in the previous embodiment, matching algorithm 18 matches the sensitivities of first and second microphone elements 14, 16, respectively, and produces first and second gain corrected signals 66, 68, respectively. First and second gain corrected signals 66, 68, respectively, are provided to adaptive algorithm 104.

Adaptive algorithm 104 combines first and second gain corrected signals 66, 68, respectively, to produce combination signal 106. In the present embodiment, adaptive algorithm 104 operates in conjunction with processor 64 of personal computer 12 to guide the direction of low sensitivity of combination signal 106. In this embodiment, adaptive algorithm 104 estimates energy levels from first and second microphone elements 14, 16, respectively. Using these energy estimates, adaptive algorithm 104 determines the location of the desired sound source, the talker, as well as the location of undesired background noise. In this fashion, the direction of low sensitivity of combination signal 106 can be directed toward the location of undesired background noise, with no restrictions on the location of that background noise, to substantially eliminate the pickup of such undesired noise. It will be appreciated by those skilled in the art that other adaptive algorithms could be utilized in the present invention. Adaptive algorithm 104 of the present embodiment represents only one example of such an adaptive algorithm.

Combination signal 106 is provided to cross-over portion 22. Cross-over portion 22 also receives first gain corrected signal 66 and first and second digital signals 56, 58, respectively, as inputs.

Referring now to FIG. 4, there is shown a block diagram of cross-over portion 22 of the embodiment of the invention depicted in FIG. 3. As shown in FIG. 4, first gain corrected signal 66 is provided to first filter portion 24, which in the present embodiment provides high frequency components from first microphone element 14. Specifically, in the present embodiment, first gain corrected signal 66 is sent to first second-order IIR filter 108. Additionally, second, third, fourth, fifth and sixth constants 110, 112, 114, 116, 118, respectively, are provided to first second-order IIR filter 108. First second-order IIR filter 108 produces first second-order IIR filter response 120.

First second-order IIR filter response 120 is sent to second second-order IIR filter 122. Additionally, seventh, eighth, ninth, tenth and eleventh constants 124, 126, 128, 130, 132,

respectively, are provided to second second-order IIR filter 122. Second second-order IIR filter 122 produces second second-order IIR filter response 134.

Second second-order IIR filter response 134 is sent to third second-order IIR filter 136. Additionally, twelfth, thirteenth, fourteenth, fifteenth and sixteenth constants 138, 140, 142, 144, 146, respectively, are provided to third second-order IIR filter 136. Third second-order IIR filter 136 produces third second-order IIR filter response 148.

Combination signal 106 is provided to second filter portion 26. In the present embodiment, second filter portion 26 is a low-pass filter. Specifically, second filter portion 26 comprises low-pass IIR filter 150. Combination signal 106 is provided to low-pass IIR filter 150. Also, seventeenth, eighteenth, nineteenth, twentieth and twenty-first constants 152, 154, 156, 158, 160, respectively, are provided to low-pass IIR filter 150. Low-pass IIR filter 150 produces low-pass IIR filter response 162. It will be appreciated by those skilled in the art that low-pass IIR filter 150 provides for either first or second-order low-pass filtering.

Third second-order IIR filter response 148 and low-pass IIR filter response 162 are each provided to first algebraic summing unit 82. First algebraic summing unit 82 unifies third second-order IIR filter response 148 and low-pass IIR filter response 162 to produce monaural signal 84. As in the prior embodiment, this embodiment results in monaural signal 84 which provides an extended frequency bandwidth. In this embodiment, however, the bandwidth is only extended at the high end because first filter portion 24 provides only high pass filtering. Although elements 86 through 102 are shown in FIG. 4, they are not discussed further here as they serve the same functions as described in relation to FIG. 1.

Although two alternative embodiments have been described herein, it will be appreciated by those skilled in the art that other embodiments of the present invention are possible. For example, other types of microphone elements could be utilized in the present invention. Specifically, first-order gradient microphone elements could be utilized to obtain a narrower pickup pattern. Additionally, various types of connectors could be utilized for first through eighth connectors 32, 34, 40, 42, 50, 52, 60, 62, respectively. For example, electrical wiring could be utilized for those connectors. It will be appreciated by those skilled in the art that the combining of signals from first and second microphone elements 14, 16, respectively, could be served by a variety of elements other than those described in the prior two embodiments. Most

simply, this combination could be accomplished by a simple algebraic subtraction accomplished by an algebraic summing unit. Furthermore, other types of filters could be utilized in place of first, second and third second-order IIR filters 108, 122, 136, respectively, and still be representative of the present invention. Moreover, it will be appreciated by those skilled in the art that first or second filter portions 24, 26, respectively, could be comprised of types of filters other than those described in connection with the above-referenced embodiments. Additionally, it will be appreciated by those skilled in the art that either or both of first and second filter portions 24, 26, respectively, could comprise one or a plurality of filter elements with a variety of functions depending upon the desired response.

Still further configurations in accordance with the present invention are possible. One such configuration comprises three or more microphones with a plurality of the microphones combined to form at least one of microphone elements 14, 16, respectively. Additionally, various combinations of hardware and/or software could be utilized in accordance with the principles of the present invention. Again, it is to be understood that the embodiments discussed herein are merely illustrative of the many possibilities of configurations which could be devised in application of the principles of the invention. Numerous other such configurations could be developed by those skilled in the art without departing from the scope of the invention.